Fresh Packet First Scheduling for Voice Traffic in Congested Networks

Maher Hamdi, Raffaele Noro and Jean-Pierre Hubaux

Abstract

We address interactive voice services over Best Effort packet networks where traffic is subject to unpredictable congestions. The quality of voice services can be significantly affected by network delay and jitter due to the rejection of late packets. LIFO has proven to be an interesting overload strategy in delay constrained systems. We propose the use of the LIFO discipline in the context of congested packet networks where voice packets are subject to delay bounds. We analyze how delay accumulates in congested queues and show that serving voice packets in LIFO improves user-level quality. An extensive simulation study shows that LIFO scheduling in congested nodes significantly reduces the fraction of late packets compared to the FIFO discipline. The enhancement of user-level quality is emphasized by using the ITU-T P.861 standard for perceptual evaluations of reconstructed voice. Finally, we show how the proposed mechanism can be easily used in the Internet.

Keywords

Interactive voice, packet networks, best effort performance, FIFO and LIFO scheduling, Internet Telephony, perceptual quality.

I. Introduction

The transport of voice over packet networks has been an active research area during the last two decades [1]. An increasing interest for voice services is stimulated by the increasing number of multimedia applications. The low cost of access to data networks, combined with PC teleconferencing tools and networked games, is likely to generate a large amount of voice communications in packet networks.

A large standardization effort has been spent to assure the interoperability of packet telephony products. In particular the ITU-T has defined the H.323 standard for visual telephone systems for networks which provide a non-guaranteed Quality of Service [2] [3]. Following this standard, an increasing number of video and audio conferencing tools are being used over Internet. More recently, to increase products compatibility the International Multimedia Teleconferencing Consortium (IMTC) approved the G.723.1 [4] codec as the low bit rate default or baseline voice coder for Internet Telephony applications based on the H.323 standard. An overview of low bit-rate speech coders is given in [5]. Another important body in this field in the Internet Telephony Consortium (ITC), a pre-standard organization that works on issues that arise from the convergence of telecommunications and the Internet.

Research aiming to improve the quality of packetized audio and video services has simultaneously evolved in two directions. Following the concept of adaptive application, the first direction focuses on optimizing the end systems (i.e. applications and terminals) to adapt them to the impairments experienced by the packet stream in the network (e.g. [6], [7], [8]). The second direction aims to provide integrated service networks with appropriate resource management mechanisms that are able to handle multiple classes of traffic and offer different qualities of service. The ATM Transfer Capacities [9] and the IETF Internet Service Model [10], [11], [12] are examples of this approach.

A large variety of queueing strategies and packet scheduling mechanisms have been proposed to be used in integrated services networks. Such mechanisms are very often a combination of buffer management (e.g., FIFO+ [13], Random Early Detection (RED) [14], Early Packet Discard (EPD) [15]), bandwidth scheduling (e.g. Generalized Processor Sharing (GPS) [16], Class-Based Queueing (CBQ) [17]) and traffic control policies (e.g. Leaky Bucket controller [9]). Some of these schemes will be used in the future Internet (in the sense of the IETF Integrated Services Group) and are even already implemented in network equipments.

In an integrated services network, one can distinguish at least two service classes: a Guaranteed Service class and a Best-Effort class. The former allows to guaranty the desired Quality of Service by means of resource reservation and admission control. Examples of this type of service are the Guaranteed Service class of the IETF Service Model [11] and the CBR and VBR Transfer Capacities of ATM networks. The Best-Effort class is illustrated by the performances of the current Internet as well as the UBR ATM Transfer Capacity. There are also intermediate classes such as IETF Controlled Load [12] and ABR ATM Transfer

Part of this work was published as: M. Hamdi, *LCFS Queuing for Real-time Audio-Visual Services in Packet Networks.* In Proceedings of International Workshop on Audio-Visual Services over Packet Networks AVSPN'97, Aberdeen, Sep. 1997.

The authors are with the Institute for computer Communications and Applications (ICA), Computer Science Department, Swiss Federal Institute of Technology (EPFL). Email: { Maher.Hamdi, Raffaele.noro, Jean-Pierre.Hubaux}@epfl.ch

EPFL Technical Report Number SSC034, December 1997

Capacity. It is commonly admitted that these classes are offered to the user on a differential pricing basis [18], the Best-Effort class being the cheapest one.

We consider the quality of voice services in a Best-Effort service class where voice packets are subject to unpredictable congestions. Despite the standardization effort and the large number of commercial and free tools for packet telephony, the main problem for voice services remains the random level of quality offered by Best Effort packet networks such as the Internet. A challenging issue, is to enhance the quality of voice over Best Effort networks. Although it is impossible to avoid congestions in such networks, we believe that Quality of Service can still be improved by adapting the effects of congestions to the specific properties of voice traffic.

In this paper we propose a scheduling mechanism that relies on the characteristics of interactive voice to minimize congestion effects and significantly enhance user-level quality even in severe load conditions. In addition to network level measurements, performance improvements are also evaluated at the user perception level. The proposed mechanism is generic and can be used for packet routers and gateways. Implementation issues in the Internet environment are also addressed in this paper.

The paper is organized as follows. In Section 2 we describe the Quality of Service requirements of a voice communication and define key performance parameters for interactive services. In Section 3 we analyze the packet delay in congested network nodes and argue why and how interactive traffic should be scheduled differently. A detailed performance evaluation of the proposed mechanism is given in Section 4. The analysis is based on simulations of network QoS parameters as well as perceptual measurements of voice communications. In Section 5 we discuss options to implement the proposed mechanism in the Internet.

II. Properties of Interactive Voice

In this section we analyze the properties of voice services in terms of QoS requirements resulting from the interactivity.

A. Real-time Services

Generally, delay sensitive communications are referred to as real-time services. For the scope of this paper, we distinguish between two types of real-time services: *delay-adaptive* services and *interactive services.*

A.1 Delay-Adaptive Services

Among real-time services, a particular attention was paid to the so-called *delay-adaptive* applications [13]. These applications tolerate short interruptions of their play-out process to adapt to packet delay variation. A typical example of delay adaptive applications is Video on Demand and more generally all media play-back services. Play-back services tolerate a few seconds delay. Generally, delay adaptive application in the Internet tolerate a end-to-end delay of few seconds.

A.2 Interactive Services

Another type of real-time applications, herein referred to as *interactive services*, are characterized by the need of a bounded packet delay in the network. These services cannot be delay-adaptive since interactivity is lost if network delay exceeds the required bound. For instance Packet Telephony, audio and video conferencing belong to this latter category. For interactive services the needed response time depends on the user requirements. We consider that the needed delay bound (denoted by D) ranges from 30 msec to 1 sec as one can reasonably assume that interactivity is lost beyond the threshold of 1 sec. Unlike delay-adaptive services, we consider that D does not change in time even if the playback point can be lower than D (for example during non congested periods).

B. Adaptation to Missing Packets

In datagram networks, packets can be lost in the network. We call *Packet Loss Ratio* (PLR) the fraction of packets that are discarded in the network. Moreover, late packets are not played out and are rejected by the receiver. This defines a *Packet Rejection Ratio, (PRR).* Depending on congestions and on the delay bound, the Rejection Ratio can be higher than the Loss Ratio. In particular, the Rejection Ratio depends on the desired response time D as it is defined by:

$$PRR = Prob[\text{Packet Delay} > D]$$

A Missing Packet is defined as a packet that has been discarded in the network or a packet that arrives too late. The *Missing Packet Ratio* MPR is than equal to

$$MPR = PLR + PRR$$

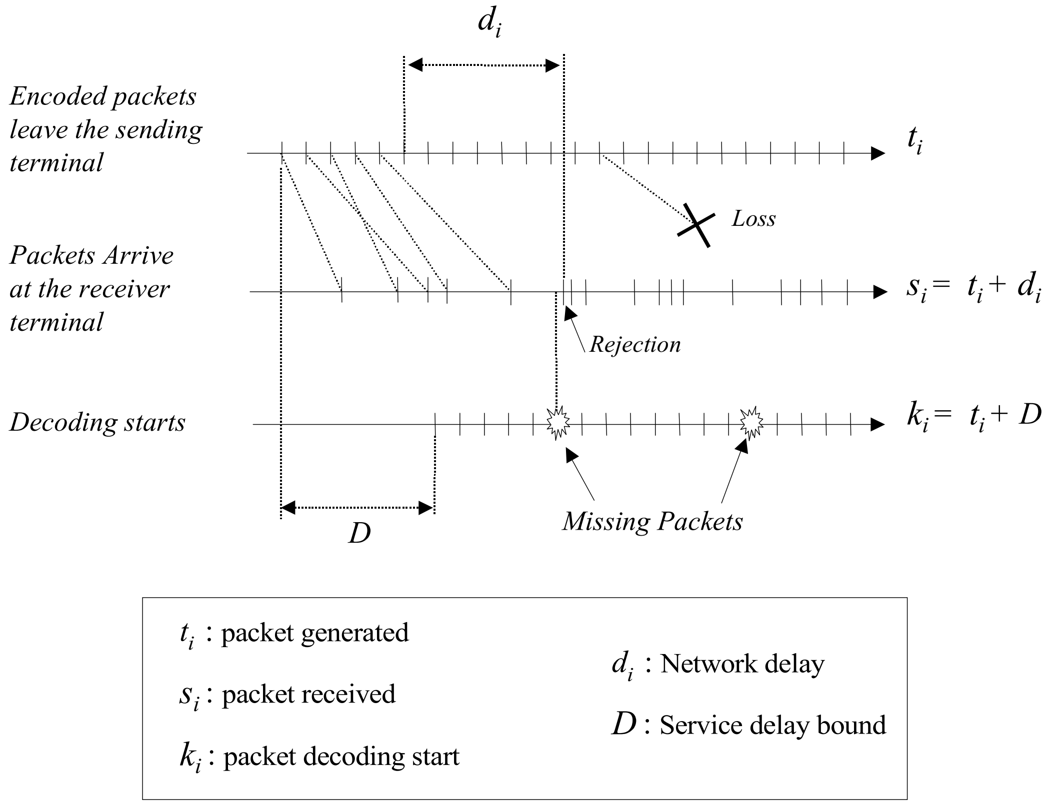

Fig. 1. Delay Diagram and Notations for Interactive Services

The delay diagram of Figure 1 illustrates the definition of interactive services considered in this paper. Following these notations, t_i is the generation instant of packet i and d_i is its end-to-end network delay. Packet i enters the decoder buffer at time $s_i = t_i + d_i$ and is consumed at $k_i = t_i + D$. Note that incoming packets are consumed in the order of their generation. If Packet i is missing in the decoder buffer at its consumption time (i.e. $s_i > k_i$), a loss recovery processing is performed.

Clearly, there is a tradeoff between interactivity (expressed by the value of D) and service quality (resulting from rejected packets). In the context of Best Effort packet networks, it is reasonably assumed that this tradeoff is more a service concern and should not have particular implications on the network.

Data carried by a missing packet is not played out and generally results in voice quality degradation. Interactive audio services, and particularly voice, are said to be *loss-adaptive* because the service is not interrupted when a packet is missing. Voice communications ability to adapt to packet loss comes from the following reasons.

- The service nature allows the listener to tolerate temporary degradations of the voice signal.
- Loss concealment techniques can be used to replace the missing data. For example a missing frame can be replaced by the last valid frame or can be interpolated by using the two adjacent frames. Concealment mechanisms for missing voice packets are described in [19] and [20].
- Forward Error Correction mechanisms consist in adding redundancy information by the transmitter. There are several FEC mechanisms proposed in the literature (e.g., [21] and [8]). The potential of FEC mechanisms to recover from losses widely depends on the packet loss process. For a given packet loss ratio, a FEC mechanism is more effective when lost packets are dispersed throughout the packet stream [7].

Unlike other communication services where the delay average and variance are significant, the performance of interactive services depends only on the fraction of missing packets. This rejection ratio is defined by the D-quantile of the packet delay distribution. It is worth noticing that all received packets are not equally useful for the decoder/player. The usefulness of a packet depends on its network delay.

III. THE *Fresh Packet First* SCHEDULING

In this section we analyze the implications of using FIFO queues in a Best Effort network on the performances of interactive services. Then we propose the Fresh Packet First scheduling as an alternative to the FIFO discipline. It should be noted that the proposed mechanism applies to interactive loss-adaptive packet services in general. However, the analysis and performance evaluations focus on the voice services.

In the analysis below we consider queues handling only interactive traffic. The implementation of the proposed mechanism in an integrated services packet network is discussed in Section 5.

A. Best-Effort Networks

We consider that a Best Effort network (or service class) has no traffic control or bandwidth reservation mechanism. Unlike Guaranteed Services class, traffic cannot be limited by using for example Connection Admission Control mechanisms. As a consequence traffic cannot be protected against long and uncontrolled congestions. These congestions can be caused by high traffic loads as well as higher priority traffic belonging to other service classes.

Voice traffic may transit through overloaded nodes either in the network (e.g., Internet routers), or in voice gateways between heterogeneous networks. The scheduling mechanism we propose addresses all bottlenecks that voice traffic may cross. Detailed examples are discussed in Section 5.

B. Packet Delay in Overloaded Queues

In network nodes, packets are generally served on a FIFO basis. In congestion periods, the node buffers starts to build up until they overflow. In a given node, it is possible to obtain bounded packets delays by limiting the buffer size and guaranteeing a minimum bandwidth. While this can be used in a guaranteed service networks, it does not scale for Best-Effort services since bandwidth cannot be guaranteed. Another reason is that the end-to-end delay depends also on the number of hops, making it impossible to set an appropriate delay bound in each node, particularly for Best-Effort networks.

We first present a simple analysis of packet delay in an overloaded FIFO queue. Consider that packet i is present in the queue at time T. We define A_i^T as the amount of time packet i has spent from its arrival in the queue until time T. We have:

$$A_i^T = T - t_i$$

where t_i denotes the arrival time of packet i to the node. A_i^T is clearly a decreasing function of t_i. When observing the FIFO service discipline from a scheduling viewpoint, it is easy to note, as stated by Clark *et al.* [13], that FIFO can be viewed as a special case of deadline scheduling where packets are served in the order of their increasing arrival times to the queues. FIFO queues are equivalent to priority queues where the highest priority matches the lowest value of packet arrival time (i.e. t_i). In other terms, the packet priority is exactly its *age*. In this context, FIFO can be referred to as an *Oldest Packet First* scheduling. One possible definition of FIFO priority (denoted by P_i^{FIFO}) is:

$$P_i^{FIFO} = A_i^T$$

It should be noted that at time T, the *delay* already spent by packet i in the queue is exactly A_i^T. With respect to the service delay bound D, priority is given to the packets that are the most likely to be rejected at the receiver. Note that this observation holds for any value of D.

A more appropriate scheduling can be drawn directly from the above analysis: packet priority should be decreasing function of packet delay:

$$P_i = -A_i^T$$

the resulting scheduling is unchanged if we add an offset T to all priorities; an equivalent assignment is:

$$P_i = t_i$$

This priority assignment leads directly to the LIFO (Last-In-First-Out) service discipline. The main idea is that LIFO gives priority to the packets that most probably will arrive before the delay bound D.

Last In First Out service discipline has been intensively studied in queueing theory (e.g [22], [23], [24], [25], [23]). The best known application of the LIFO service discipline concerns overload strategies in Electronic Switching Systems (ESS) [26]. It has also been applied to Machine Interference Problems in manufacturing systems [24].

LIFO has proven to be an interesting overload strategy when the scheduled objects (e.g., telephone calls, machine) are delay constrained. Our contribution is to make use of the LIFO discipline in the context of congested packet networks where voice packets are subject to delay bounds.

Kühn [22] analyzed the LIFO service discipline in a multi-server system with limited buffer size and derived close formulae of the waiting time distribution. Curves show that the probability of packet delay exceeding a certain value for the LIFO service discipline is orders of magnitude lower than for the FIFO discipline. This means that LIFO is expected to considerably reduce the *Packet Rejection Ratio* of interactive voice.

In fact, it is not recent knowledge that FIFO proves ineffective for real-time services (e.g. [27]). The advantage of the above analysis is to give strong intuitive arguments for the use of LIFO service discipline for interactive services in overloaded networks. Interestingly the value of D is not explicitly taken into account in this scheduling. This has the advantage that the mechanism can cope with different values of D at the same time: when implemented in a given node, the LIFO scheduling simultaneously meets the requirements of all interactive communications passing through that node. Another advantage is that, unlike explicit deadline scheduling, there is no need to partition the end-to-end delay D among all the hops of the route.

LIFO is also well known for the worst-case variance of sojourn time in single stage queues [24], [23]. Although such property can be thought of as a drawback for real-time applications, we show in the next section that it results in decorrelated missing packet occurences.

In the context of interactive services in packet networks, we refer to this mechanism as *Fresh Packet First* scheduling. It should be mentioned that we consider the non preemptive version of LIFO: a packet that started to be sent is not interrupted by the incoming packet.

Before starting the performance evaluations, it is natural to ask if there is additional complexity when using LIFO queueing in the network. Concerning network equipments (e.g., routers), handling LIFO queues has the same implementation complexity as FIFO queues: a new packet is chained at the head of the queue instead of being chained at its tail. However, one major consequence of LIFO queueing is that packets are received in an order different from their generation order. The re-sequencing operation in the receiver is dealt with in the next paragraph. The discussion of LIFO implementation in the Internet and its related protocols is presented in Section 5.

At the receiver side, out-of-sequence packets have to be re-ordered before being consumed. This means that a packet that arrives early cannot be consumed before its predecessors arrive (if ever) and are consumed. Although this may suggest that an additional delay is introduced at the receiver, it is easy to show that the delay between the source and the receiver playpoints is exactly D. Indeed, we have:

$$d_i \leq D \Longleftrightarrow s_i \leq k_i$$

This means that packet i arrives to the playout buffer before its consumption instant if and only if it has experienced a delay not greater than D. The time a packet spends in the receiver playout buffer is used for the reordering of incoming packets.

In practice, each incoming packet is inserted in its right place in the decoder buffer so that to maintain the packets sorted in the order of their sequence number. This insertion operation is an added complexity compared to FIFO queueing, and generally requires a $log(N)$ proportional computing time each packet arrival where N is the maximum number of packets in the decoder buffer[1]. It should be noted that packet mis-ordering can also happen in datagram networks (e.g IP routers) and is not exclusive to the Fresh Packet First mechanism.

IV. Performance study

In this section we present simulation results of various scenarios for congested queues. We compare the performance gain obtained when using LIFO discipline compared to the FIFO discipline. This study is two fold: in the first part we focus on network level performances such as missing packets statistics. The second part evaluates the perceptual voice quality enhancement due to the proposed scheme.

A. Simulations

We consider the generic scenario where audio traffic is mixed with a background traffic and sent through a tandem of congested nodes. The background traffic has a variable bit rate and represents other interactive services (e.g., video). Each node is represented by a single queue served at a constant rate to which the aggregate interactive traffic is offered. The service rate seen by the voice traffic is then time varying. The nodes are highly loaded and congestions occur during bursty periods of the background traffic.

[1]In practice N is no more than few tens and the insertion operation adds negligible processing delay in end systems.

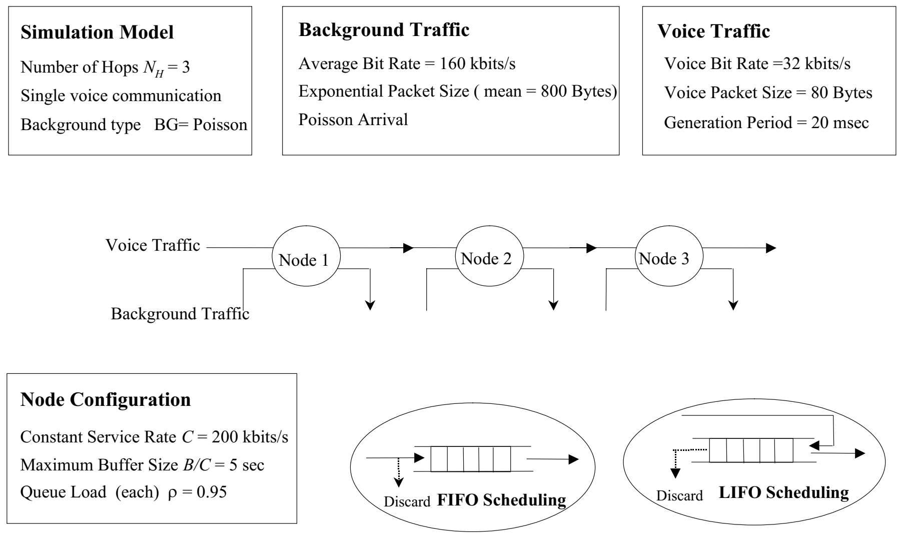

Fig. 2. Simulation Model

A.1 Simulation Model

We consider the network configuration presented in Figure 2. The traffic under observation is a voice traffic crossing a tandem of three overloaded nodes. At each node, the observed traffic is mixed with a background traffic which intensity is chosen to achieve a specified load of the node. Node buffers have limited buffer size. In the next three subsections we present simulations results when varying respectively the background traffic type, the network load, the number of hops and buffer size. The parameters setup is given by Table I.

TABLE I
Simulation models

Background Traffic		Traffic Load	Number of Hops and Buffer Size		Section
Poisson/Video	1/5 Voices	0.95	3 hops	5 sec	A.2
Poisson	1 Voice	0.5 to 1.0	3 hops	5 sec	A.3
Poisson	1 Voice	0.98	1, 3, 7 hops	0.4, 0.8, 1 sec	A.4

A.2 Background Traffic

Two kinds of background traffic have been used. The first background traffic is a Poisson arrival with exponentially distributed packet size. The second background traffic is a simple model for videoconferencing traffic given in [28]. A video packet is generated every 40 msec and its size is drawn from a first order autoregressif model. This model shows a higher correlation than that of the Poisson traffic[2]. Each node load is 0.95 and buffer delay is fixed to 5 sec. For each simulation we plot the Missing Packet Ratio (MPR) as a function of the maximum tolerated delay D.

Figure 3 shows the Missing Packet Ratio for the two types of background traffics (BG= Poisson and BG= Video). A single voice traffic is superposed to the background traffic. The two upper curves correspond to the FIFO service discipline and the two lower curves to the LIFO discipline. Due to the high network load, FIFO shows very high Missing Packet Ratio ($\tilde{9}0\%$) merely independently of the background traffic

[2]The first order autocorrelation coefficient of the video model is set to 0.8.

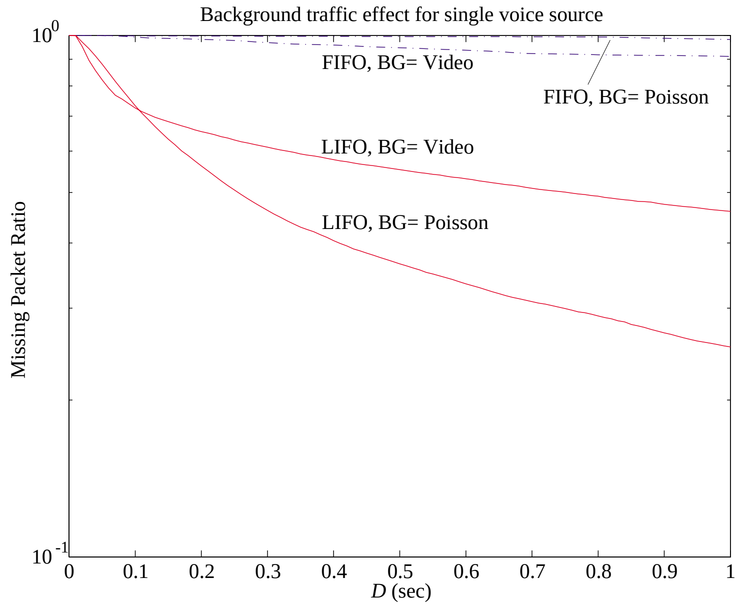

Fig. 3. Missing Packet Ratio vs D

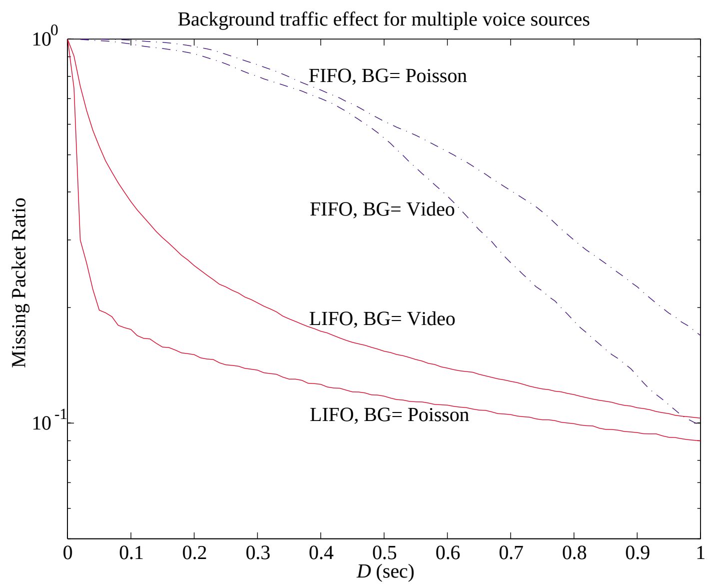

Fig. 4. Missing Packet Ratio vs D

nature. The LIFO scheduling shows much lower value for MPR than in the FIFO case; for a delay bound around 1 sec, 15 % only of voice packets are missing at the receiver, in the case of Poisson background ($\tilde{3}5\%$ for the BG = Video).

In practice, several voice connections are likely to be carried along the same path. Figure 4 shows the results for 5 voice connections multiplexed with two types of background traffics, respectively a Poisson traffic and an aggregation of 5 video sources. Again, FIFO scheduling shows higher Missing Packet Ratio than LIFO. For instance, for a delay bound of 0.5 sec and a Poisson background traffic, FIFO is responsible for 60 % of Missing Packets, five times more than in the LIFO case (12 %). It should be noted that LIFO performs better than FIFO below the threshold of 1 sec for this configuration.

A.3 Traffic Load

The goal of this set of simulations is to study the effect of network load on the performance gain. The same configuration is simulated for different load conditions. The background traffic is Poisson and a single voice traffic is considered. Figure 5 shows the Missing Packet Ratio for three value of network load : 0.5,

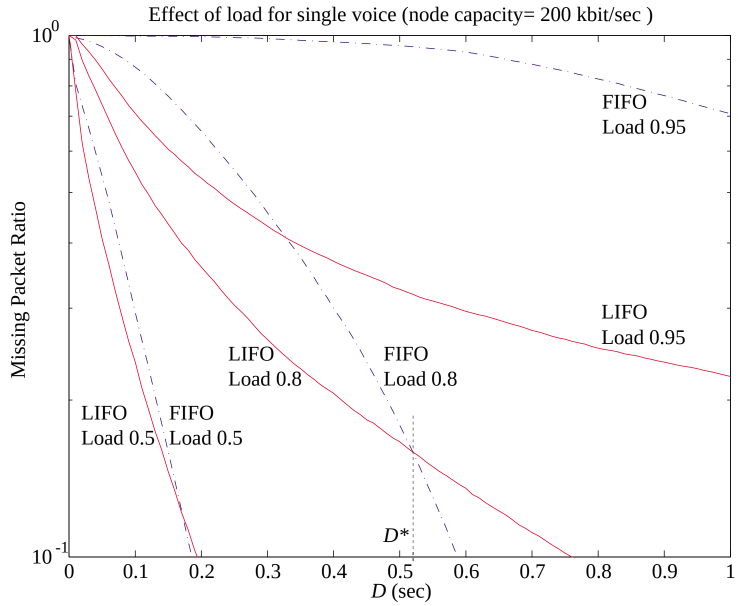

Fig. 5. Missing Packet Ratio vs D

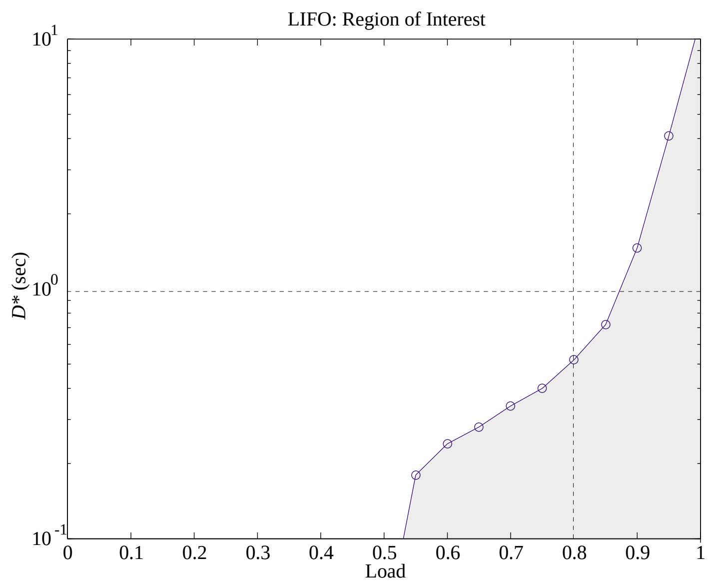

Fig. 6. LIFO Region of Interest: D^* vs ρ

0.8 and 0.95. In each case, the load value is applied to all the nodes of the path. Curves show that even for high network loads, the LIFO scheme allows a significant proportion of packets to be played out in time. If the delay bound is 0.5 sec and the network load is 0.95, the Missing Packet Ratio remains below 20% if LIFO scheduling is used while it is near 100% in the FIFO case. On the other hand, if the network load is around 0.5, curves show that both LIFO and FIFO performances remains largely compatible with interactive voice. It is clear that for reasonable values of the network load, the use of an alternative to FIFO queues is not justified. However, when dealing with frequently congested networks, the performance gained by using LIFO scheduling is clearly significant. Another observation is that LIFO is always better than FIFO below a certain value of the service delay bound (the cross point between LIFO and FIFO curves). This bound, denoted D^*, is a function of the network load and is plotted in Figure 6. In fact this function defines the region where LIFO is worth to be used: this curve upper-bounds the so-called LIFO Region of Interest. Figure 6 indeed confirms the adequacy of the Fresh Packet First scheduling to interactive services (delay bound $<$ 1sec) in congested networks (load $\tilde{0}.8$).

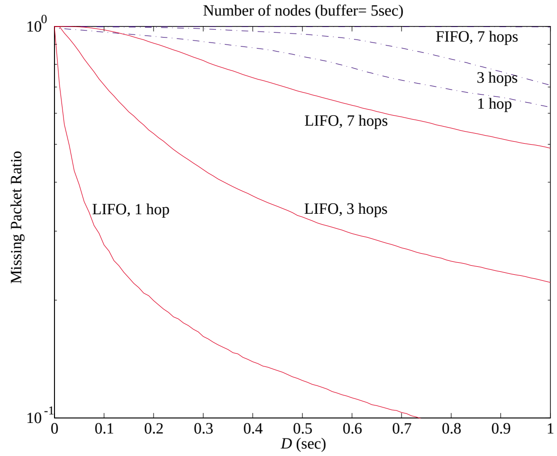

Fig. 7. Missing Packet Ratio vs D

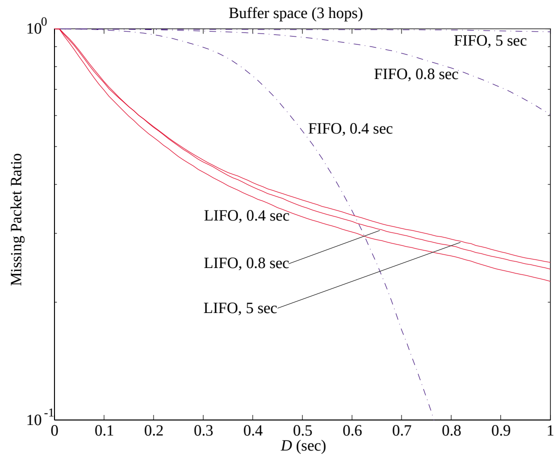

Fig. 8. Missing Packet Ratio vs D

A.4 Number of Hops and Buffer Size

The goal of this set of simulations is to study how the number of hops and buffer size influence LIFO performance gain. The offered traffic is composed by a single voice traffic and a Poisson background traffic.

The set of results presented in Figure 7 corresponds to a number of hops of 1, 3 and 7. Buffer size is fixed of 1 Mbits (i.e., 5 sec). The figure confirms that the ratio of missing packets is significantly decreased by scheduling packets in LIFO. Adding overloaded nodes to the path has an equivalent effect to increasing the queue loads. It should be noted that in practice not all the nodes crossed by a given traffic are congested at the same time. In fact the simulated configuration may represent a much longer path where the voice traffic experiences several congestions.

In Figure 8 we plot the MPR corresponding to a buffer size of 80 kbits, 160 kbits and 1 Mbits (corresponding to 0.4 sec, 0.8 sec and 5 sec respectively). The number of hops is set to 3. The main observation is that performances are much more influenced by the buffer length when FIFO is used. Clearly FIFO performs better for short buffer size because packet delays are bounded. As mentioned in Section II, it is not appropriate to enhance voice QoS by imposing a limit to the size of network buffers especially for best effort networks. It should be noted that in addition to its interesting performance, introducing LIFO scheduling in the network is a scalable way for meeting several delay requirements of interactive services.

A.5 Discussion

Results presented above cover a wide range of congestion situations in packet networks. The performance gain of the proposed mechanism mainly depends on the network load. In fact the simulated configurations reflect severe load conditions and it may be argued that this is not always the case in practice. A major feature of Best Effort networks is that there is no guaranty against congestions. Benefits of LIFO can be appreciated when considering the service availability criterion. When the *Missing Packet Ratio* reaches a

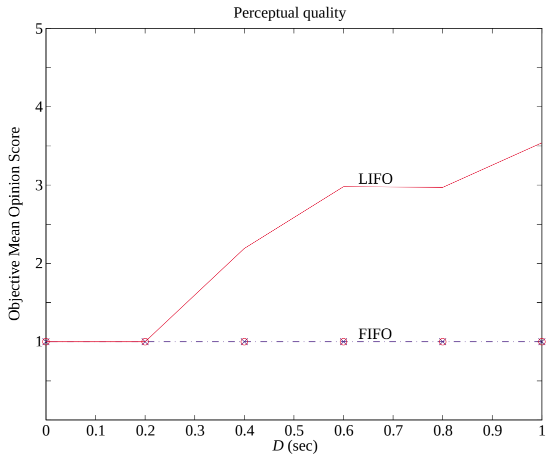

Fig. 9. Perceptual Quality Evaluation of Voice

certain threshold, the voice service is interrupted. This is caused either by the receiving end-system ability to cope with missing packets or by limits of user tolerance. The above study shows that the proposed mechanism enables voice communications to be maintained even in extreme congestions.

B. Perceptual Quality Evaluation

In this section we evaluate the performance of the Fresh Packet First mechanism from a user perception viewpoint. The reason for this perceptual evaluation is that the impact of network-level statistics on user satisfaction is generally unclear.

Carrying perceptual tests generally requires a group of people listening to speech sequences in specific conditions [29]. The most frequently used test is the absolute category rating where a five point scale is used to represent a Mean Opinion Score (MOS) for the tested sequence. The rating is given by Table II. As an example, telephone-band speech scores 4.05 in this metric.

TABLE II
Perceptual Quality Metrics

5	Excellent	degradation is imperceptible
4	Good	degradation is perceptible but not annoying
3	Fair	degradation is slightly annoying
2	Poor	degradation is annoying
1	Bad	degradation is very annoying

More recently, the ITU-T has defined the recommendation P.861 [30] for objective evaluation of speech quality. Based on this recommendation, the Perceptual Audio and Speech Quality Measure (PA&SQM^{TM}) tool [31] is used in our evaluations. The idea behind PA&SQM^{TM}to mimic sound perception of subjects as in real-life situations using a psychophysical representation of the human perception. After processing the reference and degraded signals of a speech sequence, an Objective Mean Opinion Score is given.

A hundred seconds long speech sequence [32] is used as the voice traffic under test in the simulation model specified by Figure 2. As voice decoders and players generally apply some concealment techniques to reduce packet loss effects, we used the simple "last frame substitution" technique [3] [20]. For each scheduling scheme (LIFO and FIFO) a set of degraded versions corresponding to several values of the service delay bound D is obtained. Each degraded sequence is evaluated by the PA&SQM^{TM}tool. The Objective Mean Opinion Score is plotted as a function of the service delay bound D in Figure 9. Voice quality in the FIFO case is always rated *bad*. The use of LIFO scheduling allows voice quality to be qualified as *fair* as soon a 600 msec delay is tolerated.

Keeping in mind the severe traffic conditions of the simulated network (all queues loaded to 0.95), there are two reasons why the Fresh Packet First achieves significant quality improvements. Clearly the Missing Packet Ratio is lower in the LIFO case as shown above. The second reason is due to the distribution of

[3] A missing voice frame is replaced by the last valid frame. Successive substitution are performed with an exponentially decreasing amplitude of the substituting frame.

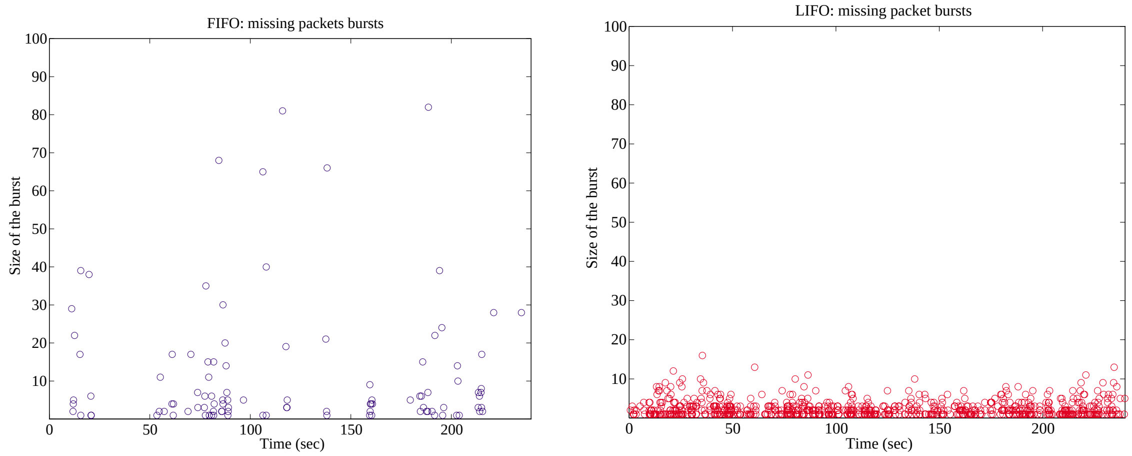

Fig. 10. Size of Missing Packet Bursts

missing packets. Correlated delays due to FIFO scheduling are responsible for successive rejected packets. When the LIFO discipline is used, the packet order in network queues is changed and delays are much less correlated than in the FIFO case. This is illustrated by Figure 10. The figures show the size of missing packet bursts as a function of time for a network load of 0.9. While the number of successive missing packets frequently reaches large values in the FIFO case, it rarely exceeds 10 when LIFO is used. This obviously increases the efficiency of missing packet concealment as well as Forward Error Correction techniques.

V. Implementation in the Internet

In this section we discuss practical issues of implementing Fresh Packet First in the Internet. We note that although the developed mechanisms consider single voice communications (packets contain voice data only), all multimedia applications that include audio are concerned because usually data, audio, video and control signals are sent in separate transport connection, as it is the case with H.323 [3] and Real-Time Protocol (RTP) [33].

A. Traffic Bottlenecks

The Fresh Packet First mechanism can be implemented in all network elements where congestions occur frequently. In this section we briefly discuss examples of network elements where the proposed mechanism can be used.

Generally speaking, congestions can be caused by a high network load. This is the case of a significant number of Internet packet routers. Moreover, the network architecture is responsible for a more specific type of congestion. It consists in permanent traffic bottlenecks that appear in equipments interconnecting heterogeneous bandwidth networks. Typical examples are LAN to WAN routing equipments used in Internet Service Providers sites (see illustration in Figure 11). Gateways used to interconnect high speed networks to low bitrate wireless networks are also subject to heavy congestion.

Fresh Packet First scheduling can also be used at a level higher than the network layer, such as multimedia gateways. Multimedia gateways are used to allow the interoperability of heterogeneous equipments such as terminals based on the H.320 to H.324 series. An interesting example is the H.323 standard that defines a gateway equipment allowing terminals connected to a best-effort packet network to interoperate with circuit-switched networks equipments. Another Telephony Gateway between packet and circuit switched networks is described in [34]. Typically such gateways are subject to congestions due to the heterogeneous bandwidths of inter-connected networks. In the remainder of this section we focus on the implementation of Fresh Packet First in the Internet.

B. Separate Buffers and Shared Bandwidth

The Fresh Packet First scheduling is designed to enhance the QoS of interactive services when the traffic is subject to hard congestions. Thus it better fits the Best-Effort class of the Internet. Note that in the Best-Effort class, bandwidth is shared among all traffics independently of their time sensitivity. Voice traffic for instance cannot be protected against bursty data traffic. However, the use of Fresh Packet First assumes that interactive traffic can be handled in separate queues than those used for non-interactive traffic. Indeed

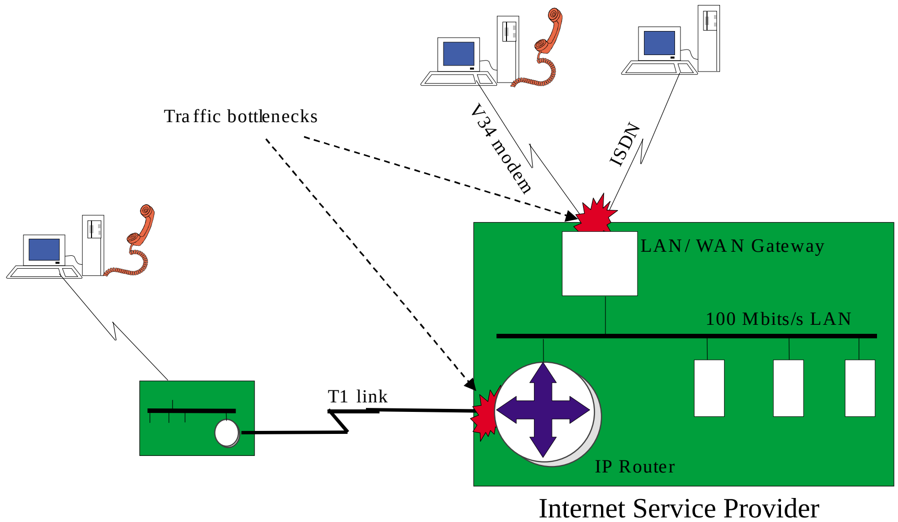

Fig. 11. Traffic Bottlenecks

the latter is not concerned with the LIFO scheduling[4]. From a scheduling point of view, interactive traffic is isolated from the others. The Fresh Packet First implementation is compatible with bandwidth sharing mechanisms proposed for Internet. We particularly consider the Weighted Fair Queueing (WFQ) [16] and the Class Based Queueing [17] mechanisms. By using the multiple queue version of WFQ [35], queues used by interactive traffic are served in LIFO. Note that the way the bandwidth is shared among all the queues remains independent from the LIFO service discipline. The Class Based Queueing allows to allocate fractions of the router bandwidth to each type of traffic [17]. Similarly, a separate queue can be used to serve interactive traffic in LIFO. Figure 12 illustrates the simplicity of implementing LIFO queues within WFQ and CBQ nodes.

The idea behind implementing Fresh Packet First in the Internet is not to serve interactive traffic better than other traffics. Indeed the proposed mechanism does not affect other traffics. We only allow interactive traffic to be served differently, in a way that matches its specific lossy and delay sensitive nature. Importantly, there is no incentive for users to declare all traffic as interactive since the only use of Fresh Packet First does not change the bandwidth available for the traffic. The Best-Effort class relies on a fair sharing of the available bandwidth. Serving voice packets in LIFO does not infringe this rule.

C. Protocols and Application Concerns

The implementation of Fresh Packet First scheduling in the Internet does not require protocol modifications. Indeed, protocol requirements of Fresh Packet First are the following:
- easy identification of packets belonging to interactive traffic in the router
- per-packet sequence numbering
- receiver compatibility with the out-of-sequence delivery of packets.

The first requirement is not specific to the Fresh Packet First implementation as it is already addressed for CBQ. Classically one may suggest to use the IP Type of Service Field[5].

The other requirements are largely supported by RTP which is actually used as a end-to-end transport protocol for Internet real-time applications. Most importantly, RTP based receivers are compatible with the misordered packet delivery since RTP is usually used on top of UDP. The implementation of Fresh Packet First does not require any modification in existing real-time IP products.

[4] It appears that data transport protocols like TCP can be dramatically disturbed if packets are systematically delivered out of order.

[5] The problem is more specific to IPv4 routers and is even easier to address in IPv6.

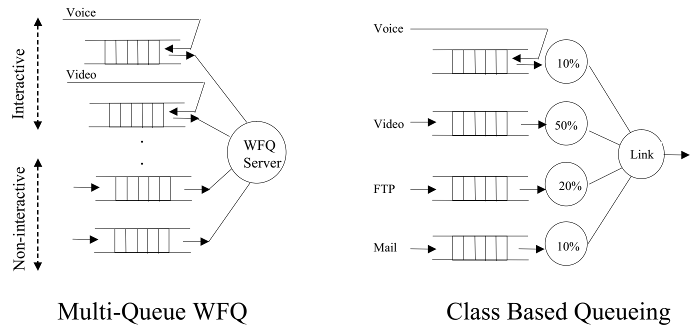

Fig. 12. Implementation with WFQ and CBQ

VI. Conclusion

In this paper we describe a scheduling mechanism for interactive voice services in packet networks. The mechanism is based on using the LIFO discipline in overloaded queues. A large set of network simulations have been performed under different traffic conditions. The resulting traffic profiles have been used as input for a perceptual quality evaluation software. Results have demonstrated the quality improvement gained by using LIFO scheduling.

The proposed mechanism is very simple to implement in the Internet. Indeed, the only systems affected are the routers, whereas the end-systems do not require any modification. The change of the scheduling policy can be implemented for instance by taking advantage of a new release of the router's software.

It is interesting to consider the combination of Fresh Packet First performances with other approaches that aim to enhance the quality of packet audio services. The use a Forward Error Correction mechanism, based on the statistics of missing packets when the Fresh Packet First scheduling is used, is also promising. Although our work was focused on voice services, we expect a similar impact on interactive video services.

VII. Acknowledgments

The authors are indebted to Gerard Hebuterne, Laurent Toutain, Jean-Yves Le Boudec, Paul Kühn and Patrick Thiran, for very interesting discussions. The PA&SQM^{TM}software was kindly provided by OPTICOM.

References

[1] C. Weinstein and J. Forgie, "Experience With Speech Communication in Packet Networks", *IEEE Journal on Selected Areas on Communications*, vol. 1, pp. 963–980, Dec. 1983.
[2] G.A. Thom, "H.323: The Multimedia Communications Standard for Local Area Networks", *IEEE Communications Magazine*, vol. 34, pp. 52–56, Dec. 1996.
[3] ITU-T Rec. H.323, "Visual Telephone Systems and Terminal Equipement fr Local Area Networks which Provide a Non-Guaranteed Quality of Service", 1996.
[4] ITU-T Rec. H.723.1, "Dual Rate Speech Codec for Multimedia Telecommunications Transmitting at 6.4 and 5.3 kbits/s", 1995.
[5] R.V. Cox and P. Kroon, "Low Bit-Rate Speech Coders for Multimedia Communication", *IEEE Communications Magazine*, vol. 34, pp. 34–41, Dec. 1996.
[6] J.C. Bolot, T. Turletti, and I. Wakeman, "Scalable Feedback Control for Multicast Video Distribution in the Internet", *in Proceedings of SigComm*. ACM, Sep. 1994.
[7] J.C. Bolot and A. Vega-Garcia, "Control Mechanisms for Packet Audio in the Internet", *in Proceedings of Infocom'96*. IEEE, Mar. 1996.
[8] N. Shacham and P. McKenney, "Packet Recovery in High-Speed Networks using Coding and Buffer Management", *in Proceedings of INFOCOM'90*. IEEE, May 1990.
[9] ATM Forum, "UNI Specifications V4".
[10] S. Shenker and J. Wroclawski, "Network Element QoS Control Service Specification Template, RFC 2216", Sep. 1997.
[11] S. Shenker, C. Partridge, and R. Guerin, "Specification of the Guaranteed Quality of Service, RFC 2212", Sep. 1997.
[12] J. Wroclawski, "Specification of the Controlled Load Quality of Service, RFC 2211", Sep. 1997.

[13] D. Clark, S. Shenker, and L. Zhang, "Supporting Real-Time Applications in an Integrated Services Packet Network: Architecture and Mechanism", *in Proceedings of SIGCOMM*. ACM, Aug. 1992.
[14] S. Floyd and V. Jacobson, "Random Early Detection Gateways for Congestion Avoidance", *IEEE/ACM Transactions on Networking*, vol. 1, pp. 397–413, Aug. 1993.
[15] A. Romanow and S. Floyd, "Dynamics of TCP Traffic over ATM Networks", *IEEE JSAC*, vol. 13, 1995.
[16] A.K. Parekh and R.G. Gallager, "A Generalized Processor Sharing Approch to Flow Control In Intergated Services Networks-The Single Node Case", *IEEE/ACM Transactions on Networking*, vol. 1, pp. 344–357, June 1993.
[17] S. Floyd and V. Jacobson, "Link Sharing and Resource Management Models for Packet Networks", *IEEE/ACM Transactions on Networking*, vol. 3, 1995.
[18] S. Shenker, "Service Models and Pricing Policies for an Integrated Services Internet", *Public Access to the Internet, B. Kahin and J. Kedler eds., MIT Press*, 1993.
[19] D.J. Goodman, J.B. Lockhart, O.J. Wasem, and W. Wong, "Wavefom Substitution Techniques for Recovering Missing Speech Segments in Packet Voice Communications", *IEEE Trans. On Acoustics, Speech and Signal Processing*, vol. 34, pp. 1440–1448, Dec. 1986.
[20] K. Yokota, T. Suzuki, O. Noguchi, and Y. Shoji, "A New Missing ATM Cell Reconstruction Scheme For ADPCM-Coded Speech", *in Proceedings of GLOBECOM'89*, pp. 1926–1930. IEEE, 1989.
[21] J.C. Bolot and A. Vega-Garcia, "The Case for FEC-Based Error Control for Packet Audio in the Internet", *in To appear in ACM Multimedia System*. ACM, 1997.
[22] P. Kühn, "On a Combined Delay and Loss System with Different Queue Discipline", *in Transactions of the Sixth Prague Conference on Information Theory, Statistical Decision Functions and Random Processes, Prague*, Sep. 1971.
[23] L. Durr, "A Single Server Priority Queueing System with General Holding Times, Poisson Input and Reverse-Order-of-Arrival Queueing Discipline", *Operations Research*, vol. 17, 1969.
[24] M. Mittler and R. Muller, "Sojourn Time Distribution of the Asymmetric M/M/1/N - System with LCFS-PR Service", *University of Wuerzburg-Institute of Computer Science Research Report Series. Report No. 94*, 1994.
[25] S.S. Panwar, D. Towsley, and Jack K. Wolf, "Optimal scheduling policies for a class of queues with customer deadlines to the beginning of service", *Journal of the ACM*, vol. 35, pp. 832–844, Oct. 1988.
[26] L.J. Forys, "Performance Analysis of a New Overload Strategy", *in Proceedings of the 10th International Teletraffic Congress, Montreal*, 1983.
[27] R. Chipalkatti, J. Kurose, and D. Towsley, "Scheduling Policies for Real-Time and Non-Real-Time Traffic in a Statistical Multiplexer", *in Proceedings of INFOCOM'89*, pp. 774–783. IEEE, 1989.
[28] D. P. Heyman, A. Tabatabai, and T. V. Lakshman, "Statistical Analysis and Simulation Study of Video Teleconference Traffic in ATM Networks", *IEEE Transactions on Circuits and Systems for Video Technology*, vol. 2, pp. 49–59, Mar. 1992.
[29] ITU-T Rec. P.80, "Telephone Transmission Quality Subjective Opinion Tests", 1994.
[30] ITU-T Rec. P.861, "Objective Quality Measurement of Telephone-Band (300-3400 Hz) speech codecs", 1996.
[31] OPTICOM, "PA&SQM (TM): Perceptual Audio and Speech Quality Measure V6.0", *Product of OPTICOM (www.opticom.de). Copyrighted by KPN Resarch, The Netherlands.*, 1997.
[32] S.L. Jackson and J. Travolta, "Royale With Cheese", *Dialogue from the Motion Picture: Pulp Fiction. MCA Records, Inc*, 1994.
[33] H. Schulzrinne, S. Casner, R. Frederick, and V. Jacobson, "RTP A Transport Protocol for Real-Time Applications, RFC 1889", Jan. 1996.
[34] P. L. Simeonov, P. Hofmann, M. Rebensburg, and D. Ruffler, "@INGate: A Distributed Intelligent Network Approach to Bridge Switching and Packet Networks", *in Proceedings of ICCCN'97*. IEEE, Sep. 1997.
[35] J.C.R Bennet and H. Zhang, "WF2Q: Worst-Case Fair Weighted Fair Queueing", *in Proceedings of INFOCOM'96*, pp. 120–128. IEEE, Mar. 1996.